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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
10/057,981	01/29/2002	Gagan Choudhury	003493.00294	2528
28317	7590	12/09/2005	EXAMINER	
BANNER & WITCOFF LTD., COUNCEL FOR AT& T CORP. 1001 G STREET , N.W. ELEVENTH STREET WASHINGTON, DC 20001-4597			WONG, WARNER	
			ART UNIT	PAPER NUMBER
			2668	
DATE MAILED: 12/09/2005				

Please find below and/or attached an Office communication concerning this application or proceeding.

<b>Office Action Summary</b>	Application No.	(K)	Applicant(s)
	10/057,981		CHOWDHURY ET AL.
	Examiner Warner Wong	Art Unit 2668	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --  
Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

#### Status

- 1) Responsive to communication(s) filed on 29 January 2002.
- 2a) This action is FINAL.                    2b) This action is non-final.
- 3) Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

#### Disposition of Claims

- 4) Claim(s) 1-38 is/are pending in the application.
  - 4a) Of the above claim(s) \_\_\_\_\_ is/are withdrawn from consideration.
- 5) Claim(s) 37 and 38 is/are allowed.
- 6) Claim(s) 1-36 is/are rejected.
- 7) Claim(s) \_\_\_\_\_ is/are objected to.
- 8) Claim(s) \_\_\_\_\_ are subject to restriction and/or election requirement.

#### Application Papers

- 9) The specification is objected to by the Examiner.
- 10) The drawing(s) filed on 29 January 2002 is/are: a) accepted or b) objected to by the Examiner.  
Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).  
Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).
- 11) The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

#### Priority under 35 U.S.C. § 119

- 12) Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).
    - a) All    b) Some \* c) None of:
      1. Certified copies of the priority documents have been received.
      2. Certified copies of the priority documents have been received in Application No. \_\_\_\_\_.
      3. Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).
- \* See the attached detailed Office action for a list of the certified copies not received.

#### Attachment(s)

- |  |  |
|--|--|
| 1) <input checked="" type="checkbox"/> Notice of References Cited (PTO-892)  | 4) <input type="checkbox"/> Interview Summary (PTO-413)<br>Paper No(s)/Mail Date. _____. |
| 2) <input type="checkbox"/> Notice of Draftsperson's Patent Drawing Review (PTO-948)   | 5) <input type="checkbox"/> Notice of Informal Patent Application (PTO-152)              |
| 3) <input checked="" type="checkbox"/> Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08)<br>Paper No(s)/Mail Date _____. | 6) <input type="checkbox"/> Other: _____.  |

**DETAILED ACTION*****Claim Objections***

The following claims are objected to because of the following informalities:

1. Claim 9, line 1: The limitation "measured voice-path delay" lacks antecedent basis. It appears that claim 9 might be dependent on claim 8 instead.
2. Claim 9, line 2: The limitation "the predetermined value of the voice-path delay" lacks antecedent basis. It appears that claim 9 might be dependent on claim 8 instead.
3. Claim 26, line 5: The limitation "a predetermined maximum value" appears to be the same as that specified in claim 26, lines 2-3). It should be changed to "the predetermined maximum value".
4. Claim 27, line 4: The limitation "a predetermined minimum value" appears to be the same as that specified in claim 27, lines 2-3). It should be changed to "the predetermined minimum value".

Appropriate action is required.

***Claim Rejections - 35 USC § 102***

5. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

A person shall be entitled to a patent unless –

(e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

6. Claims 1-2 and 8-11 are rejected under 35 U.S.C. 102(e) as being anticipated by Shaffer (6,683,889).

**Regarding claim 1,** Shaffer describes a method for managing the depth of a (de)-jitter buffer comprising:

determining an initial depth parameter (fig. 6, #550), rate of change parameter (condition) for changing the depth value (fig. 6, condition states #560 & #562 as the parameter/condition of the varying amount in buffer “min. unplayed” & “max. unplayed” leading to changing the depth value [fig. 6, #561 & 564 for increasing/decreasing buffer size] ), threshold depth parameter for the de-jitter buffer (fig. 6, #552);

monitoring a characteristic of the de-jitter buffer (fig. 6, #556);

modifying the initial depth parameter and threshold depth parameters based on the monitoring (fig. 6, #558, #561, #564).

**Regarding claim 2,** Shaffer describes all limitations set forth in claim 1. Shaffer further describes that the monitoring comprises comparing a characteristic of the de-jitter buffer with a predetermined value of the characteristic (fig. 6, #560, #562).

**Regarding claim 8,** Shaffer describes all limitations set forth in claim 1. Shaffer further describes the monitoring comprises comparing a measured voice-path delay (Min. Unplayed [unprocessed/delayed packets]) with a predetermined value of the voice-path delay (T1 and/or T2) (fig. 6, #560, #562).

**Regarding claim 9,** Shaffer describes all limitations set forth in claim 1. Shaffer further describes the measured voice-path delay is greater than or equal to the predetermined value of the voice-path delay (fig. 6, #562).

**Regarding claim 10,** Shaffer describes all limitations set forth in claim 9. Shaffer further describes the modifying comprises modifying parameters (buffer depth/size) such that the voice-path delay is reduced (col. 3, lines 13-18).

**Regarding claim 11,** Shaffer describes all limitations set forth in claim 1. Shaffer further describes the monitoring comprises comparing a measured voice-path delay (Min. Unplayed [unprocessed/delayed packets]) with a predetermined value of the voice-path delay (T1 and/or T2) the measured voice-path delay being less than the predetermined value of voice-path delay (fig. 6, #560).

***Claim Rejections - 35 USC § 103***

7. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

(a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negated by the manner in which the invention was made.

8. Claims 3-7 and 12 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer in view of Hatano (5,737,314).

**Regarding claim 3,** Shaffer describes all limitations set forth in claim 2. Shaffer lacks what Hatano describes:

the characteristic is a measured packet/cell loss probability and the predetermined value of the characteristic is a predetermined (acceptable) value of the measured packet/cell loss [probability] (col. 8, lines 34-42).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use the packet/cell loss probability values used by the system of Hatano for monitoring (and controlling) the depth of the de-jitter buffer of Shaffer. The motivation being that the revised method would guarantee (and conform to) packet/cell loss probability stipulated by ITU recommendation I.371 (Hatano, col 1. lines 10-14 and lines 60-63), yielding it to be compatible for use in ITU I.371 compliant networks.

**Regarding claim 4,** Shaffer and Hatano combined describe all limitations set forth in claim 3. Hatano further describes:

The measure packet loss probability is greater than or equal to the predetermined value of the packet loss probability (Hatano, col. 8, lines 37-42, & abstract, "an overflow testing circuit .. to decide, according to the distribution, whether an event has occurred that the cell loss probability is shifted due to the input cells to a distribution related to a cell loss probability exceeding an upper-limit value of acceptable cell loss probability").

**Regarding claim 5,** Shaffer and Hatano combined describe all limitations set forth in claim 4. Hatano further describes:

modifying parameters such that the packet/cell loss probability is reduced (col. 30, lines 42-45, & abstract, "An ATM switch system having a traffic control function to reduce cell loss probability".

**Regarding claim 6,** Shaffer describes all limitations set forth in claim 2. Shaffer lacks what Hatano describes:

the monitoring comprises comparing a measured packet/cell loss probability with a predetermined packet/cell loss probability value and determining that the measured packet/cell loss probability is less than the predetermined value of the packet/cell loss probability (col. 8, lines 42-47, & abstract, "an underflow testing circuit for testing , according to the distribution, whether an event has occurred that the cell loss probability is shifted to a distribution related to a cell loss probability less than a lower-limit value of acceptable cell loss probability").

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use the packet/cell loss probability values used by the system of Hatano for monitoring (and controlling) the depth of the de-jitter buffer of Shaffer. The motivation being that the revised method would guarantee (and conform to) packet/cell loss probability stipulated by ITU recommendation I.371 (Hatano, col 1. lines 10-14 and lines 60-63), yielding it to be compatible for use in ITU I.371 compliant networks.

**Regarding claim 7**, Shaffer and Hatano combined describe all limitations set forth in claim 6. Hatano further describes:

modifying parameters such that the voice-path (cell) delay is reduced (col. 8, lines 42-47, & abstract, "An ATM switch system having a traffic control function to reduce cell loss probability and delay of cells".

**Regarding claim 12**, Shaffer describes all limitations set forth in claim 11.

Shaffer lacks what Hantano describes:

the modifying comprises modifying parameters (buffer depth/size) such that the packet loss probability is reduced (col. 30, lines 42-45, & abstract, "An ATM switch system having a traffic control function to reduce cell loss probability".

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use the packet/cell loss probability values used by the system of Hatano for modifying the depth of the de-jitter buffer of Shaffer. The motivation being that the revised method would guarantee (and conform to) packet/cell loss probability stipulated by ITU recommendation I.371 (Hatano, col 1. lines 10-14 and lines 60-63), yielding it to be compatible for use in ITU I.371 compliant networks.

Claim 13 is rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer in view of Amster (6,961,315).

9. Shaffer describe all limitations set forth in claim 1. Shaffer lacks what Amster describes: the (monitoring) of an R-factor (call quality) (col. 2, lines 21-23, "To monitor voice quality, at least one probe packet is injected for transport across the transmission path.", where call quality is measured by a R-factor score, "Thus, there is a need for a technique that enables a fully analytical measurement of the R factor in terms of measureable transport metrics..", col. 2, lines 9-11.)

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to monitor the call quality/R-factor. The motivation is "to facilitate ongoing measurements of voice quality without apriori knowledge of the equipment within the network", (Amster, col. 2, lines 11-13).

10. Claims 14-15 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer in view of Amster as applied to claim 13 above, and further in view of Yarroll (2003/0115320).

**Regarding claim 14,** Shaffer and Amster combined describe all limitations set forth in claim 13. Shaffer and Amster fail what Yarroll describes:

the modifying comprises modifying parameters (jb(0) & pbr) such that the R-factor is increased (paragraph 26).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to increase the R-factor/call quality in the buffer de-jittering method of Shaffer and Amster. The motivation being that the modified method would then “account for cognitive effects such as listener perception and the effects on a loss on the [network] coder-decoder.” (Yarroll, paragraph 3, “However, the method of the ‘538 patent does not account for cognitive effects such as listener perception and the effects of loss on the coder-decoder (CODEC).”)

**Regarding claim 15,** Shaffer, Amster and Yarroll combined describe all limitations set forth in claim 14. Amster further describe that R-Factors measures voice quality (col. 1, lines 65-66).

11. Claims 16-17 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer in view of Bharucha (2003/0064187).

**Regarding claim 16,** Shaffer describes all limitations set forth in claim 1. Shaffer lacks what Bharucha describes: classifying the incoming call (paragraph 37, where the signal classifier classifies the type of call).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to classify the incoming calls de-jittering buffer method of Shaffer. The motivation being that “the ATM switch (network device with the method) may better able to deal with congestion situations where the ATM switch is supplied with information about the call type from the signal classifier 21 for each of the active VCs”, Bharucha, paragraph 40.

**Regarding claim 17,** Shaffer and Bharucha combined Shaffer describes all limitations set forth in claim 1. Bharucha furthers describe that the classifying comprises determining characteristics (bandwidth) of the incoming call and grouping the call into a category based on the determined characteristics (paragraph 37, “For example, bandwidth of the virtual circuit may depend on the type of call being initiated as, for example, detected at the signal classifier 21 with different bandwidths utilized for different type of calls.”)

12. Claims 18 –24 are rejected under 35 U.S.C. 103(a) as being unpatentable over Shaffer in view of Bharucha as applied to claim 18 above, and further in view of Kuroshita (5,550,807), Jones (2002/0161542) and Hallenstal (2001/0036158).

**Regarding claim 18,** Shaffer and Bharucha combined describes all limitations set forth in claims 17. Bharucha further describes the determined characteristics may

be from the type of calls (paragraph 37, where the signal classifier classifies the type of call).

Shaffer and Bharucha combined lack what Kuroshita describes: the determined characteristics may be from the (physical) distance between a transmitter (equipment 11) and a receiver (i-th equipment) (col. 6, lines 34-39).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use the physical distance as a determined characteristic for monitoring the incoming call of the de-jittering method of Shaffer and Bharucha. The motivation for this is because the physical distance is directly proportional to the delay of the transmitted data in which the de-jittering method is managing.

Shaffer, Bharucha and Kuroshita combined lack what Jones described: the determined characteristics may be from the type of access (paragraph 9, "The communication channel [type of access] may comprise any channel including fiber optic cable, coaxial cable, power transmission line, network line Ethernet, twisted pair or any channel capable of conducting data.")

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use access type as a determined characteristic for monitoring/controlling the transmission. The motivation being that "The medium must be considered because the medium and its condition can affect the rate at which communication may occur" (Jones, paragraph 2).

Shaffer, Bharucha, Kuroshita and Jones lack what Hallenstal describes: the determined characteristics may be from the type of backbone (paragraph 73, "It should

also be understood that the term "broadband", as used herein and in appended claims, embraces and encompasses packet-switched technologies in general [backbone types] (e.g. IP, VoIP, Fram-relay, ATM, etc.") or egress/terminal type (terminal capability) (paragraph 69, "The PSTN/ISDN node 330(2) then selects a B-channel which can be used to reach terminal 324(D-P) or negotiates with the terminal 324(D-P) as which B-channel to use [depending on the terminal type and the protocol type ISDN or PSTN].")

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use the backbone type or egress/terminal type (capability) mentioned in Hallenstal as a determined characteristic for monitoring/controlling the transmission. The motivation being that each backbone type and terminal type may be used by a particular connection/transmission which affects the rate and delay of the transmission.

**Regarding claim 19**, Shaffer, Bharucha and Kuroshita, Jones and Hallenstal combined describe all limitations set forth in 18. Bharucha further describes the type of call being determined is being selected from the group consisting voice, fax and voice-band data (paragraph 30, "For example, the signal classifier (SC) 21 may be configured to identify voice, fax and voice-and data calls.")

**Regarding claim 20**, Shaffer, Bharucha and Kuroshita, Jones and Hallenstal combined describe all limitations set forth in 18.

Bharucha further describes the type of access [of voice calls] may be from standard local loop (line from a telephone at subscriber's premises to the telephone company central office switch (fig. 1, telephone #2 connecting to toll switch #5).

Jones further describes the type of access [of the voice calls] may be from wire cable pair (twisted pair), coaxial cable, wireless (radio frequency waveguide) and Ethernet (paragraph 41, "The line to be probed may comprise any type of conductor or channel including, but not limited to, a twisted pair conductor, coaxial cable, Ethernet, an optic channel, or a radio frequency waveguide.", which also may cover DSL medium, paragraph 3).

**Regarding claims 21-22,** Shaffer, Bharucha and Kuroshita, Jones and Hallenstal combined describe all limitations set forth in 18.

Hallenstal further describes: determining the type of backbone [utilized by the voice calls] may be from Internet Protocol (IP, Asynchronous Transfer Mode (ATM) and Frame Relay (paragraph 73, "It should also be understood that the term "broadband", as used herein and in appended claims, embraces and encompasses packet-switched technologies in general (e.g. IP, VoIP, Fram-relay, ATM, etc.)") as well as egress/terminal type (terminal capability) [of the voice calls] (paragraph 69, "The PSTN/ISDN node 330(2) then selects a B-channel which can be used to reach terminal 324(D-P) or negotiates with the terminal 324(D-P) as which B-channel to use [depending on the terminal type and the protocol type ISDN or PSTN].")

**Regarding claim 23,** Shaffer, Bharucha, Kuroshita, Jones and Hallenstal combined describe all limitations set forth in 18.

Kuroshita further describes: determining the physical distance between a transmitter (equipment 11) and a receiver (i-th equipment) (col. 6, lines 34-39).

**Regarding claim 24,** Shaffer, Bharucha, Kuroshita, Jones and Hallenstal

combined describe all limitations set forth in 23. Kuroshita further describes:

transmitting a signal to a receiver at a first point in time (time at which the request packet 31 has been transmitted), detecting a response from the receiver at a second point in time (time at which the response packet 32 is received), measuring the difference between the first point in time and the second point in time (computing the time interval) and correlating the difference with the physical distance (col. 6, lines 34-39).

13. Claims 25-27 are rejected under 35 U.S.C. 102(e) as being anticipated by Pate (6,757,292).

**Regarding claim 25,** Shaffer describes a method for managing the depth of a (de)-jitter buffer comprising:

receiving a data packet associated with an incoming [VoIP] call, said data packet being associated with a delay (PDV) (col. 1, lines 55-59);  
increasing the de-jitter buffer depth (JAB size) by a predetermined first percentage (table 2, where buffer size D(M) is changed to D(H) in transitioning from M/MR state to H/HR state) if the delay (PDV(p)) is greater than a first threshold value (threshold M, col. 6, lines 14-15) of the depth of the de-jitter buffer (col. 5, lines 27-28).

decreasing the de-jitter buffer depth (JAB size) by a predetermined second percentage (table 2, where the buffer size D(H) is changed to D(M) in transitioning from state H/HR state to M/MR state) if the delay (PDV(p)) is less than a second threshold

value (threshold L, col. 6, line 10-11) of the depth of the de-jitter buffer (col. 5, lines 29-32).

**Regarding claim 26,** Pate describes all limitations set forth in claim 25. Pate further describes:

increasing the de-jitter buffer depth by a predetermined percentage comprises setting the de-jitter buffer depth to a predetermined maximum value (D(H)) if the delay is greater than the first threshold value of the depth of the de-jitter buffer and if the depth of the de-jitter buffer is (already) equal to the predetermined maximum value (D(H)) (table 2, where H and HR states both have the jitter buffer depth D(H) when transitioning from H to HR to re-center the JAB by modifying the output clock rate [instead of the jitter buffer depth], col. 5, lines 24-25, "If the JAB is almost empty or almost full then re-center the JAB by modifying the output clock rate").

**Regarding claim 27,** Pate describes all limitations set forth in claim 25. Pate further describes:

decreasing the de-jitter buffer depth by a predetermined second percentage comprises setting the depth of the de-jitter buffer to a predetermined minimum value (D(L)) if the delay is less than the second threshold value of the depth of the de-jitter buffer and if the depth of the de-jitter buffer is less than a predetermined minimum value (D(L)) (table 2, where L and LR states both have the jitter buffer depth (D(L)) when transitioning from L to LR to re-center the JAB by modifying the output clock rate [instead of the jitter buffer depth], col. 5, lines 24-25, "If the JAB is almost empty or almost full then re-center the JAB by modifying the output clock rate").)

14. Claim 28 is rejected under 35 U.S.C. 103(a) as being unpatentable over Pate in view of Bharucha (2003/0064187).

Pate describes all limitations set forth in claim 25. Pate lacks what Bharucha describes: classifying the incoming call (paragraph 37, where the signal classifier classifies the type of call).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to classify the incoming calls to the de-jittering buffer method of Pate. The motivation being that “the ATM switch (network device with the method) may better able to deal with congestion situations where the ATM switch is supplied with information about the call type from the signal classifier 21 for each of the active VCs”, Bharucha, paragraph 40.

15. Claim 29-32 is rejected under 35 U.S.C. 103(a) as being unpatentable over Pate in view of Kim (2002/0172148).

**Regarding claims 29,** Pate describes a method for managing depth of a de-jitter buffer comprising:

receiving a data packet associated with an incoming [VoIP] call, said data packet being associated with a delay (PDV) (col. 1, lines 55-59);

increasing the de-jitter buffer depth (JAB size) by a predetermined first percentage (table 2, where buffer size D(M) is changed to D(H) in transitioning from

M/MR state to H/HR state) if the delay (PDV(p)) is greater than a first threshold value (threshold M, col. 6, lines 14-15) of the depth of the de-jitter buffer (col. 5, lines 27-28); increasing the de-jitter buffer depth by a predetermined second percentage (table 2, where buffer size D(L) is changed to D(M) in transitioning from L/LR state to M/MR state) (col. 5, lines 27-28); decreasing the de-jitter buffer depth (JAB size) by a predetermined third percentage (table 2, where the buffer size D(M) is changed to D(L) in transitioning from state M state to L/LR state) (col. 5, lines 29-32).

Pate lack what Kim describes:

[where increasing buffer depth by the second percentage], if the delay is less than the first threshold value (Delta C – highest threshold) and greater than or equal to a second threshold (Delta 2 – medium threshold) value, the first threshold value being greater than the second threshold value (fig. 6, #242, 244,246 and paragraph 69, with 3 delay thresholds where Delta C > Delta 2 > Delta 1 in terms of delay/congestion.)

[where decreasing de-jitter buffer depth by the third percentage], if the delay is less than the second threshold (Delta 2 – medium threshold) and greater than or equal to a third threshold value (Delta 1 – lowest threshold), the second threshold value being greater than the third threshold value.

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use a multi-level thresholds as in Kim to evaluate and enact upon a packet delay to further alter the buffer depth in Pate. The motivation being that the modified method will [better] handle congestion, “[Therefore], congestion control

methods .. involved methods that relay on modifying or shaping the traffic at the traffic source,.. , in order to prevent or to handle congestion.", Kim, paragraph 5.

**Regarding claim 30,** Pate and Kim combined describe all limitations set forth in claim 29. Pate further describes the increasing of the de-jitter buffer depth by a predetermined first percentage comprises setting the de-jitter buffer depth to a predetermined maximum value (D(H)) (table 2, where M state transitioning to MR or H state changes the buffer depth to D(H)).

**Regarding claim 31,** Pate and Kim combined describe all limitations set forth in claim 29. Pate further describes the increasing of the de-jitter buffer depth by a predetermined second percentage comprises setting the de-jitter buffer depth to a predetermined (maximum) value (D(M)) (table 2, where L state transitioning to LR or M state changes the buffer depth to D(M)).

**Regarding claim 32,** Pate and Kim combined describe all limitations set forth in claim 29. Pate further describes the decreasing of the de-jitter buffer depth by a predetermined third percentage comprises setting the de-jitter buffer depth to a predetermined (minimum) value (D(L)) (table 2, where M state transitioning to LR or L state changes the buffer depth to D(L)).

16. **Claim 33** is rejected under 35 U.S.C. 103(a) as being unpatentable over Pate in view of Kim, and further in view of Ren (6,456,590).

Pate and Kim combined describe all limitations set forth in claim 29. Pate and Kim lacks what Ren describes:

Setting the depth of the de-jitter buffer equal to a predetermined initial value only if the data packet is the first packet of the incoming call (i.e. static [one-time] allocation) (abstract, "The static scheme allocates the shared memory 50 evenly among the input ports 20 or based on the input port [incoming call] transmission rate.")

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use a static allocation scheme of Ren in allocating the de-jittering buffer size of each incoming call. The motivation being that utilizing a static allocation scheme "maximize system simplicity and efficiency", (Ren, col. 4, lines 9-11).

17. **Claim 34** is rejected under 35 U.S.C. 103(a) as being unpatentable over Pate in view of Kim, and further in view of Bharucha (2003/0064187).

Pate and Kim combined describe all limitations set forth in claim 29. Pate and Kim lacks what Bharucha describes:

classifying the incoming call (paragraph 37, where the signal classifier classifies the type of call).

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to classify the incoming calls to the de-jittering buffer method of Pate and Kim. The motivation being that "the ATM switch (network device with the method) may better able to deal with congestion situations where the ATM switch is supplied with information about the call type from the signal classifier 21 for each of the active VCs", Bharucha, paragraph 40.

18. Claim 35 is rejected under 35 U.S.C. 103(a) as being unpatentable over Pate in view of Bharucha (2003/0064187).

**Regarding claims 35,** Pate describes a method for managing depth of a de-jitter buffer, where the buffer has an ideal (initial) buffer depth (table 2, L (Low) initial state with Jitter Adaptation Buffer JAB depth set to D(L)), the method comprising:

receiving a data packet associated with an incoming [VoIP] call, said data packet being associated with a delay (PDV) (col. 1, lines 55-59);

Pate lacks what Bharucha describes:

setting the depth of the de-jitter buffer equal to the ideal (initial) buffer depth only if the data packet is the first packet of the incoming call (i.e. static [one-time] allocation) (abstract, "The static scheme allocates the shared memory 50 evenly among the input ports 20 or based on the input port [incoming call] transmission rate.")

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to use a static allocation scheme of Ren in allocating the de-jittering buffer size of each incoming call. The motivation being that utilizing a static allocation scheme "maximize system simplicity and efficiency", (Ren, col. 4, lines 9-11).

19. **Claim 36** is rejected under 35 U.S.C. 103(a) as being unpatentable over Larson (4,569,042) in view Pate and further in view of Sato (5,646,568).

Larson describes as a prior art a method for calculating a delay, comprising:  
receiving a data packet associated with an incoming call, computing an actual arrival instant of the data packet and a reference zero-delay arrival instant of the data

packet (zero delay reference), calculating a delay based on said actual arrival instant of the data packet and the reference zero-delay arrival instant of the data packet (col. 2 lines 6-18).

Larson further describes:

dropping the data packet if the delay if the actual arrival instant of the data packet is greater than the reference zero-delay arrival instant of the data packet (col. 4, lines 22-25).

Larson lacks what Pate describes:

(delay compensating) depth of a de-jitter buffer [to evaluate whether to drop the data packet [if the delay is greater than the buffer depth]] (table 2, depths D(L), D(M) and D(H));

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to include the (delay compensating) depth of a de-jitter buffer in evaluating of whether or not to drop the data packet. The motivation being that if the delay of the packet exceeds the (delay compensating) depth of the de-jittering buffer, the functionality of the de-jittering buffer fails since it cannot sustain the delay of the (incoming) packet.

Larson and Pate lacks what Sato describes:

adjusting a minimum delay value (upward) [if the delay if the actual arrival instant of the data packet is greater than the reference zero-delay arrival instant of the data packet] (col. 3, lines 31-33, "In the delay circuit of FIG. 1, the fixed delay element 12a is

provided to adjust a minimum delay time which is a period of time by which a pulse signal arrives at selector 11b.”)

It would have been obvious to one with ordinary skill in the art at the time of invention by applicant to adjust the minimum delay value. The motivation being that the adjustment may yield, “As a result, a period corresponding to the maximum delay time of the delay circuit of FIG.1 as a whole should coincide with a maximum delay period of the input pulse signal 31.”)

***Allowable Subject Matter***

20. Claims 37-38 allowed.

***Conclusion***

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Warner Wong whose telephone number is 571-272-8197. The examiner can normally be reached on 5:30AM - 2:00PM, M-F.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Chieh Fan can be reached on 571-272-3042. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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Warner Wong  
Examiner  
Art Unit 2668

WW

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